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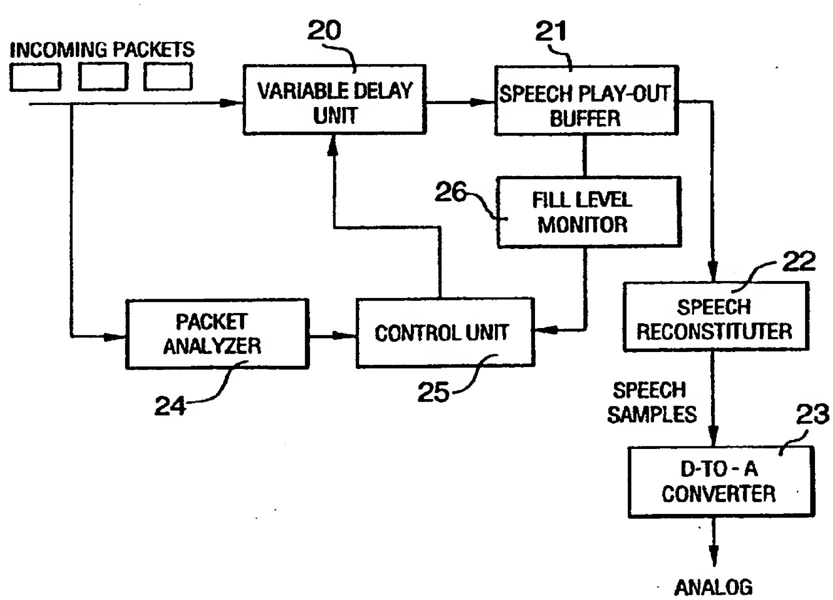

(57) Abstract

A method is described for playing out packets, such as voice or video packets, received through a packet network subject to variable transmission delays. The incoming packets are received in a delay buffer and a predetermined delay applied to the first packet of a sequence of packets. A variable delay is applied to subsequent packets to produce an appropriate constant play-out rate to reproduce the desired output. The fill level of the delay buffer is monitored and the predetermined delay applied to the first packet of a following sequence of packets adjusted to maintain the fill level within desired limits to minimize the risk of said buffer underflowing or overflowing.

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METHOD OF DYNAMICALLY COMPENSATING FOR VARIABLE TRANSMISSION DELAYS IN PACKET NETWORKS

This invention relates to a method and apparatus for dynamically compensating for variable transmission delays in packet networks, particularly voice networks, but the invention is also applicable to other networks, such as video networks. The invention is applicable in all integrated packet networks where voice or video may be carried including, for example, frame relay networks, ATM networks, PCME (packet circuit multiplication equipment), and LANs.

In this specification, reference is made throughout to "voice" packets, since this is the term normally used in the art to describe such networks, although it will be realized by one skilled in the art that such networks extend to any network capable of transmitting any form of audio whether it actually be voice or other form of reproducible sound.

Two methods have been proposed to compensate for variable transmission delays in packet voice networks.

In the first method, known as timestamping, which is used in ITU standard G.764, the accumulated variable transmission delay experienced by a voice packet is recorded in a timestamp field in the packet. Each intermediate node recognizes voice packets and adds to the timestamp field the amount of time that it took for the packet to transit the node. The receiver uses the value in the timestamp field to determine when to play-out the voice packet. Voice packets that experience little delay in the network will be delayed in the receiver longer before being played-out, and voice packets that experienced long network delays will be delayed less in the receiver. The effect is that the sum of the network delay and the delay in the receiver will

be nearly constant for all voice packets and the voice will be played-out at a uniform rate.

The disadvantage of this method is that intermediate nodes must recognize voice packets and carry out special processing. This makes this method incompatible with existing networks that do not support this function. Another disadvantage is that prior knowledge of the maximum expected delay variation is necessary.

In the second method, known as the blind delay method, a fixed delay is always added at the receiver to the first packet of a talk sequence. The delay corresponds to the maximum variable delay expected from the network. This way, if the first packet experiences minimum delay, the system compensates by adding enough delay to make sure other packets, which experience more delay, arrive before their scheduled play-out time.

The disadvantage of this method is that it may increase the delay in the voice path beyond the optimal value. This is because if first packet has already experienced the worst case delay, the delay added will include the worst case twice. Large delays degrade the system performance (or may cause the system not to meet the international standards for network delays specified in ITU-T Recommendation G.114). Delay is of special concern when speech and facsimile demodulated traffic are mixed on the same transmission facility. Another disadvantage is that prior knowledge of the maximum expected delay variation is necessary.

According to the present invention there is provided a method of playing out packets received through a packet network subject to variable transmission delays, comprising the steps of receiving incoming packets in a buffer; applying a delay to the first packet of a sequence of packets; applying a variable delay to subsequent packets of the sequence to produce an

appropriate constant play-out rate to reproduce the desired output; monitoring utilization of said buffer; and adjusting the delay applied to the first packet of a following sequence of packets to maintain said utilization within desired limits to minimize the risk of said buffer underflowing or overflowing.

The utilization monitored can be, for example, the buffer fill level or the dwell time of packets in the buffer. Alternatively, the arrival rate of the packets could be monitored.

The packets may, for example, be voice packets or video packets.

The invention (Adaptive Delay Equalization) thus uses an adaptive method to determine when to play-out received packets. This method minimizes the delay applied to received packets. In a preferred embodiment, the receiver starts by applying a pre-determined delay to the first packet of a talk-spurt. The receiver delays subsequent packets by an amount appropriate to produce a constant gap-free play-out rate. If the minimum number of packets in the buffer is large (i.e., the buffer is never close of under-flowing), the system slowly reduces the predetermined delay, known as the build-out delay. If packets arrive late, the build-out delay is increased in order to minimize packet loss. The build-out delay adjustment can be done during speech silence. The duration of gaps between spoken words is precisely replicated by sending a 'silence duration' value in the first packet of each new talk spurt.

This method has the advantage that it minimizes the delay when the network is not congested, adapts itself to operate optimally in various network conditions, without any user reconfiguration, and it does not require any

complicated and specialized node handling of voice packets (e.g. time-stamping).

The invention also provides an apparatus for playing out packets received through a packet network subject to variable transmission delays, comprising a buffer for receiving incoming packets; a speech reconstituter for receiving said packets from said buffer and reconstituting speech samples therefrom; a variable delay unit for a applying a delay to the incoming packets; a control unit for controlling said delay to apply a first delay to an incoming sequence of packets and a variable delay to subsequent packets of the sequence so as to produce an appropriate constant play-out rate to said buffer; means for monitoring the utilization of said buffer; and means for adjusting said first delay applied to the packets of a following sequence of packets to maintain said fill level within desired limits to minimize the risk of said buffer underflowing or overflowing.

The invention will now be described in more detail, by way of example only, with reference to the accompanying drawings, in which:-

Figure 1 is a schematic diagram showing a variable transmission delay packet voice network ;

Figure 2 is a timing diagram of a packetized voice transmission system in accordance with the invention; and

Figure 3 is a block diagram of a variable delay compensation apparatus in accordance with the invention.

As shown in Figure 1, a speech input is transmitted in packetized form through a packet network 2, for example an ATM or Frame Relay network, which introduces a variable delay during transmission through the network.

The incoming packets are received by receiver 3, which outputs a re-assembled speech signal.

The packet network 2 introduces a variable propagation delay Δ. Receiver 3 introduces a further delay δ in the manner to be described.

Referring now to Figure 2, input speech consists of spurts 4 separated by periods of silence 5. Each spurt 4 is represented by a sequence of packets 6, which when they are transmitted are separated by fixed spaces 7 as shown at line 8. However, after transmission through the network the packets are no longer equally spaced, as shown at line 9, due to the variable propagation delays in the network. In accordance with the invention, as shown at line 10, the first packet 6a of each sequence is subjected to a predetermined delay, which is estimated to be adequate to avoid buffer underflow and overflow. The remaining packets of the sequence are subjected to variable delays to maintain the appropriate constant output 11. This is then decoded to reproduce the initial speech as shown at line 12.

Referring now to Figure 3, the incoming speech packets 6 are fed to variable delay unit 20, which introduces a variable delay between the packets. The output of variable delay unit 20 is fed to speech play-out buffer 21, which outputs the speech packets to speech reconstituter 22, which turns the speech packets into constant rate speech samples, normally at 8KHz. These speech samples are then converted into analog speech signals in digital-to-analog converter 23.

Incoming speech packets 6 are also fed to packet analyzer 24 whose function is to identify the start of a speech spurt and trigger control unit 25, which sets the delay introduced by the variable delay unit 20 so as to produce a constant output rate. Buffer fill level

monitor 26 monitors the fill level of speech play-out buffer 21. Depending on the fill level of buffer 21, control unit 25 varies the initial delay for the start of the next talk spurt. Monitor 26 can be replaced by a similar unit monitoring the dwell time of the packets in the buffer. Alternatively, the buffer utilization can be determined by monitoring the arrival rate of the packets.

In operation, the control unit 25 applies a pre-determined delay to the first packet of a talk-spurt detected by packet analyzer 24. The control unit 25 then delays subsequent packets by an amount appropriate to produce a constant play-out rate to the speech play-out buffer 21. If the minimum number of packets in the buffer is large (i.e., the buffer never becomes close to under-flowing), the control unit 25 slowly reduces the build-out delay. If packets arrive late, (i.e., the buffer risks under-flowing), the build-out delay is increased in order to minimize packet loss. Adjustment of the build-out delay can be determined in a number of ways, such as monitoring the minimum, maximum or average utilization of the buffer. Alternatively, it is possible to monitor the time a packet spends in the buffer.

Preferably, the delay adjustment is done during speech silence. Normally, the duration of gaps between spoken words is precisely replicated by sending a silence duration value in the first packet of each new talk spurt, which can be detected by the packet analyzer 24. When the build-out delay has to be increased, the silence duration is artificially increased. The result is a larger build-out delay during the next talk spurt. When the build-out delay has to be decreased, the silence duration is artificially decreased. The result is a shorter build-out delay during the next talk spurt.

Although the preferred embodiment has been described with reference to audio signal, the invention may also be

applied to video transmission. In this case video packets carry the video data, and the speech reconstituter is replaced by a video reconstituter, which operates in an analogous manner. Indeed the invention is applicable to any digitized physical signal that is transmitted in packetized format and then reconstituted at the far end. The video implementation looks the same as the implementation shown in the drawings with the word "video" substituted for the word "speech" throughout.

Claims:

1. A method of playing out packets received through a packet network subject to variable transmission delays, characterized in that it comprises the steps of:

a) receiving incoming packets in a buffer;

b) applying a delay to the first packet of a sequence of packets;

c) applying a variable delay to subsequent packets of the sequence to produce an appropriate constant play-out rate of said packets to reproduce the desired output;

d) monitoring the utilization of said buffer; and

e) adjusting the delay applied to the first packet of a following sequence of packets to maintain said buffer utilization within desired limits to minimize the risk of said buffer underflowing or overflowing.

2. A method as claimed in claim 1, characterized in that in step d the fill level of said buffer is monitored.

3. A method as claimed in claim 1, characterized in that in step d the dwell time of said packets in said buffer is monitored.

4. A method as claimed in claim 1, characterized in that said packets are voice or audio packets.

5. A method as claimed in any of claims 1 to 4, characterized in that each said sequence represents a signal spurt.

6. A method as claimed in claim 5, characterized in that said delay adjustment is carried out during signal activity in the gaps between signal spurts.

7. A method as claimed in claim 5, characterized in that a signal inactivity duration value is inserted in the first packet of each spurt.

8. A method as claimed in claim 1, characterized in that said packets are video packets.

9. An apparatus for playing out packets received through a network subject to variable transmission delays, comprising:

a) a buffer for receiving incoming packets;

b) a signal reconstituter for receiving said packets from said buffer and reconstituting signal samples therefrom;

c) a variable delay unit for applying a delay to the incoming packets;

d) a control unit for controlling said delay to apply a first delay to an incoming sequence of packets and a variable delay to subsequent packets of the sequence so as to produce an appropriate constant play-out rate to said buffer;

e) means for monitoring the utilization of said buffer; and

f) means for adjusting said first delay applied to the packets of a following sequence of packets to maintain said buffer utilization within desired limits to minimize the risk of said buffer underflowing or overflowing.

10. An apparatus as claimed in claim 9, characterized in that said means for monitoring the utilization of said buffer comprises a buffer fill level monitor.

11. An apparatus as claimed in claim 9, characterized in that said means for monitoring the utilization of said buffer comprises a monitor determining the amount of time the packets spend in the buffer.

12. An apparatus as claimed in claim 9, characterized in that it further comprises a packet analyzer for detecting the start of a signal spurt.

13. An apparatus as claimed in claim 7, characterized in that said delay adjustment is carried out during the gaps between signals to be transmitted.

14. An apparatus as claimed in claim 9, characterized in that said packets are voice or audio packets.

15. An apparatus as claimed in claim 9, characterized in that said packets are video packets.

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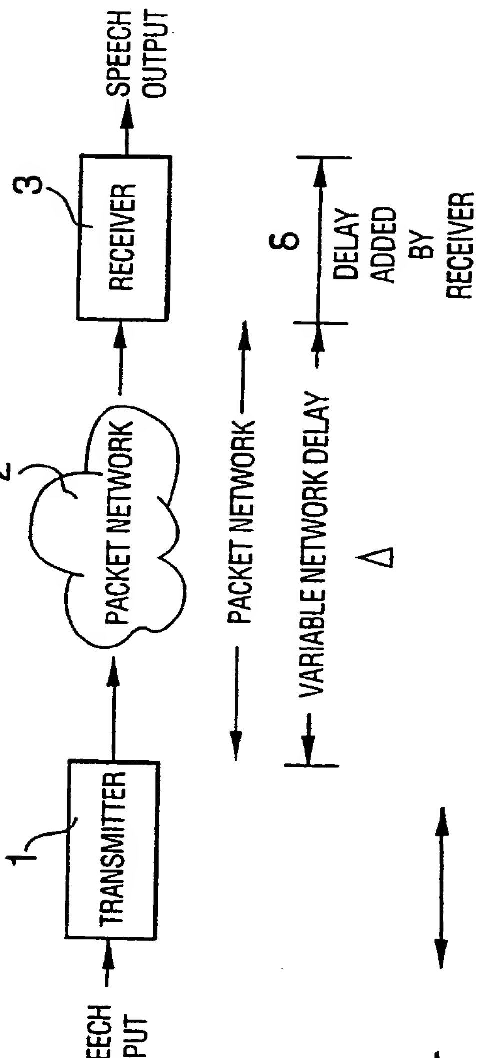
SPEECH INPUT
1
TRANSMITTER
2
PACKET NETWORK
3
RECEIVER
SPEECH OUTPUT
PACKET NETWORK
VARIABLE NETWORK DELAY
Δ
δ
DELAY ADDED BY RECEIVER

FIG.1

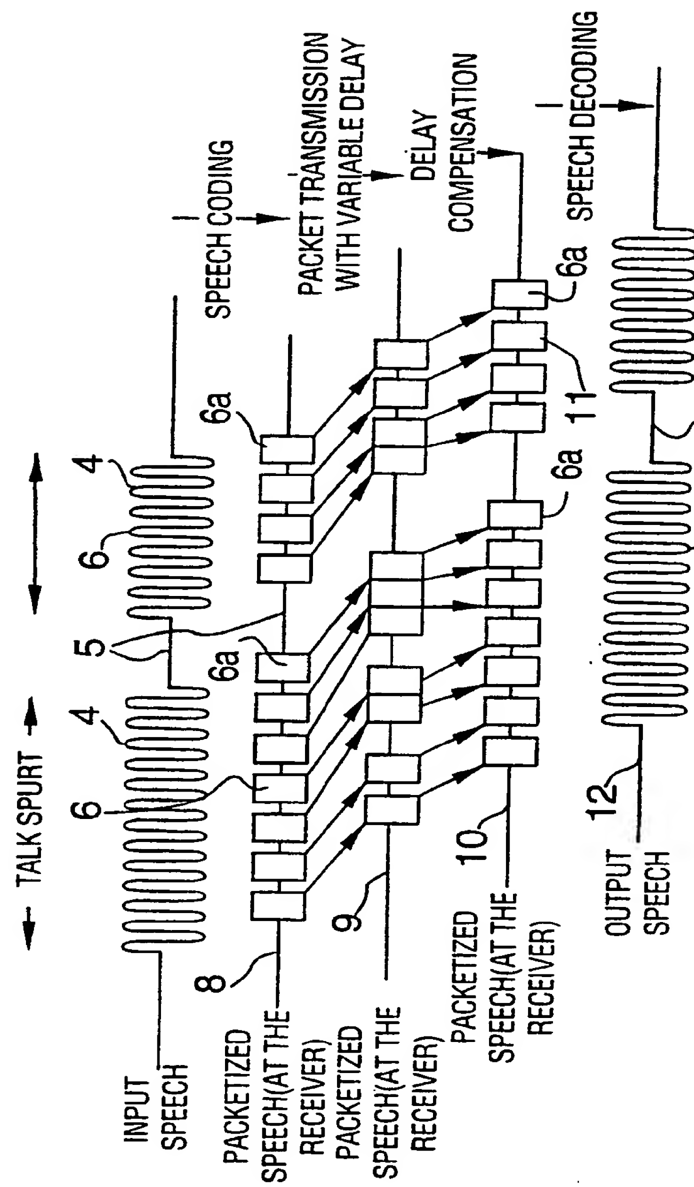
TALK SPURT
INPUT SPEECH
SPEECH CODING
PACKETIZED SPEECH(AT THE RECEIVER)
PACKET TRANSMISSION WITH VARIABLE DELAY
PACKETIZED SPEECH(AT THE RECEIVER)
DELAY COMPENSATION
PACKETIZED SPEECH(AT THE RECEIVER)
SPEECH DECODING
OUTPUT SPEECH
4
5
6
6a
8
9
10
11
12

FIG.2

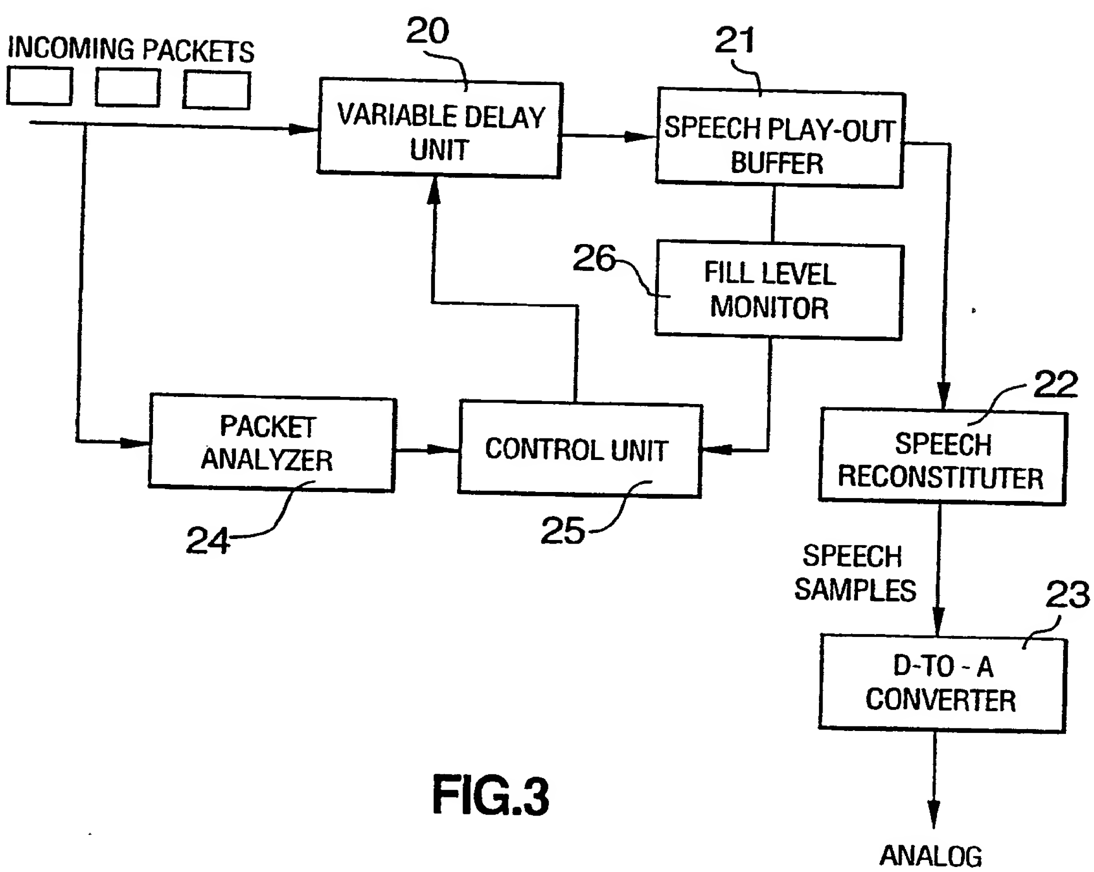
INCOMING PACKETS
20
VARIABLE DELAY UNIT
21
SPEECH PLAY-OUT BUFFER
26
FILL LEVEL MONITOR
22
SPEECH RECONSTITUTER
PACKET ANALYZER
24
CONTROL UNIT
25
SPEECH SAMPLES
23
D-TO - A CONVERTER
ANALOG

FIG.3

INTERNATIONAL SEARCH REPORT

Inte nal Application No
PCT/CA 95/00062

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04Q11/04 H04J3/06

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04Q H04J H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS, DEC. 1983, USA, vol. SAC-1,no. 6, December 1983 ISSN 0733-8716, pages 1022-1028, MONTGOMERY W A 'Techniques for packet voice synchronization' see page 1022, right column, paragraph 3 - page 1023, left column, paragraph 4	1-6,8
A	see page 1023, right column, paragraph 7 - page 1024, left column, paragraph 1 see page 1024, right column, paragraph 3 see page 1026, left column, paragraph 6 - page 1027, right column, paragraph 1; figures 1,5,6 ---	9-15

-/--

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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Date of the actual completion of the international search	Date of mailing of the international search report
18 May 1995	08. 06. 95

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Patent document cited in search report	Publication date	Patent family member(s)		Publication date
EP-A-130431	09-01-85	US-A-	4538259	27-08-85
		JP-C-	1728158	19-01-93
		JP-B-	4011059	27-02-92
		JP-A-	60027255	12-02-85
WO-A-8701254	26-02-87	AU-B-	604650	03-01-91
		AU-A-	6330986	10-03-87
		AU-B-	635805	01-04-93
		AU-A-	6937791	11-04-91
		CA-A-	1273697	04-09-90
		CA-A-	1334685	07-03-95
		DE-A-	3688595	22-07-93
		EP-A,B	0235257	09-09-87
		JP-T-	63500697	10-03-88
		KR-B-	9400395	19-01-94
		US-A-	4782485	01-11-88
		US-A-	5018136	21-05-91

INTERNATIONAL SEARCH REPORT

Inter ...al Application No
PCT/CA 95/00062

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP,A,0 130 431 (IBM) 9 January 1985 see page 11, line 16 - page 12, line 7 see page 14, line 31 - page 16, line 20; figures 2,3 ---	1,2,4,5, 9,10,14
A	WO,A,87 01254 (REPUBLIC TELCOM SYSTEMS) 26 February 1987 see page 7, line 6 - line 11 see page 19, line 31 - page 21, line 1; figure 4 ---	1,9
A	EBU REVIEW, TECHNICAL, JUNE 1991, BELGIUM, no. 247, June 1991 ISSN 0379-7155, pages 124-131, XP 000228454 ASSMUS U ET AL 'High-quality video and audio signal transmission in a broadband ISDN based on ATD' see abstract see page 128, paragraph 6.1; figure 6 -----	1,2,4,5, 8-10,14, 15

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